User's Guide

Hammerfall® DSP System

Multiface

The most compact professional multitrack recording system ever!

TotalMix™

SyncAlign® ZLM® SyncCheck®

PCI Busmaster Digital I/O System
PCI and CardBus Interface
2 + 24 Channels Stereo / ADAT Interface
24 Bit / 96 kHz Analog Stereo Monitor
ADAT Sync In

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1. Introduction

Thank you for choosing the Hammerfall DSP. This unique audio system is capable of transferring digital audio data directly to a computer from practically any device equipped with a digital audio interface, be it S/PDIF, AES/EBU or ADAT optical. Installation is simple, even for the inexperienced user, thanks to the latest Plug and Play technology and full interrupt-sharing. The numerous unique features and well thought-out configuration dialog puts the Hammerfall DSP at the very top of the range of digital audio interface cards.

The package includes drivers for Windows 98/2000/XP and MacOS. An ALSA driver for Linux is planned to be available soon (see chapter 7.4).

Our high-performance philosophy guarantees maximum system performance by executing all functions directly in hardware and not in the driver (i.e. the CPU).

2. Package Contents

Please check that your Hammerfall DSP System's package contains each of the following:

PCI Interface:
- PCI card HDSP
- Quick Info guide
- RME Driver CD
- Cable IEEE1394, 4.5 m (15 ft)
- Internal cable (3 pin)

CardBus Interface:
- CardBus card
- Quick Info guide
- RME Driver CD
- Cable CardBus to IEEE1394, 4.5 m (15 ft)
- 12 V car cable
- Battery cable
- Power supply 12 V / 1.25 A and power cord

Multiface:
- I/O-box Multiface
- Quick Info guide
- RME Driver CD
- 1 optical cable (TOSLINK), 2 m (6.6 ft)

3. System Requirements

- Windows 98/SE/ME, Windows 2000/XP, Linux, MacOS 9.x
- PCI Interface: a free PCI rev. 2.1 Busmaster slot
- CardBus Interface: a free PCMCIA Slot type II, CardBus-compatible

Note: Examples and detailed descriptions of suitable audio desktop systems can be found in the Tech Info *RME Reference PCs: Hardware recommendations*. Information on compatibility and performance of notebooks/laptops is included in RMEs Tech Infos about notebooks, *HDSP System – Notebook Basics and Tests*.

4. Brief Description and Characteristics

- Hammerfall design: 0% (zero!) CPU load, even using all 36 ASIO channels
- All settings can be changed in real-time
- Enhanced mixed mode: ADAT In, S/PDIF In, and all outputs can be used simultaneously
- 8 available buffer sizes/latencies: 1.5 / 3 / 6 / 12 / 23 / 46 / 93 / 186 ms
- Sample Split technology for 4 channel, 96 kHz/24-bit record/playback via ADAT optical
- Slave and master clock modes
- Automatic and intelligent master/slave clock control
- Unsurpassed Bitclock PLL (audio synchronization) in ADAT mode
- Word clock input and output
- ADAT Sync in (9-pin D-type) for sample-accurate transfer
- Zero Latency Monitoring: Hardware bypass per track, controlled by Punch in/out
- Enhanced ZLM for latency-free submixes and perfect ASIO Direct Monitoring
- SyncAlign guarantees sample aligned and never swapping channels
- SyncCheck tests and reports the synchronization status of input signals
- 1 x MIDI I/O, 16 channels high-speed MIDI
- 1 x Analog Line/headphone output, separate output for independent submix
- DIGICheck DSP: Level meter in hardware, peak- and RMS calculation
- TotalMix: 720 channel mixer with 40 bit internal resolution

5. Technical Specifications

5.1 Digital

- Super Low Jitter Design: < 3 ns word clock PLL, < 2 ns ADAT PLL, < 1 ns internal
- Internal sample rates: 32 / 44.1 / 48 / 88.2 / 96 kHz
- Supported sample rats through word clock: 27 kHz - 103 kHz
- Internal resolution: 24 Bit
- Input PLL ensures zero dropout, even at more than 40 ns jitter
- Bitclock PLL for trouble-free varispeed operation in ADAT mode
- High-sensitivity input stage (< 0.2 Vss input level)
- Output voltage 0.8V (consumer mode, phono) or 2.3V (professional mode)
- Phono input and output ground-free transformer coupled
- Connectors: optical (TOSLINK), phono, BNC
- Clocks: ADAT Sync In, word clock I/O
- Formats: SPDIF (Consumer and Professional), ADAT optical

5.2 Analog

Stereo Monitor Output

- Analog output level: +8 dBu @ 0 dBFS
- Dynamic range: 108 dB (RMS unweighted, unmuted), 112 dBA
- THD+N: -100 dB / 0.001%
- Frequency response DA, -0.1 dB: 20 Hz - 20.8 kHz (sf 44,1 kHz)
- Frequency response DA, -0.5 dB: 10 Hz - 44 kHz (sf 96 kHz)
- Sample rates playback: 32 / 44.1 / 48 / 64 / 88.2 / 96 kHz and variable (word clock)
- Ouput impedance: 75 Ohm
- Channel separation: > 110 dB

AD

- Resolution AD: 24 Bit
- Signal to Noise ratio: 101 dB RMS unweighted, 106 dBA
- THD: < -107 dB, < 0.00045 %
- THD+N: < -96 dB, < 0.0016 %
- Crosstalk: > 120 dB
- Analog headroom prior to AD conversion: 13 dB
- Frequency response AD @ 44.1 kHz, -0.5 dB: 5 Hz - 20.7 kHz
- Frequency response AD @ 96 kHz, -0.5 dB: 5 Hz - 32 kHz
- Input Line: 1/4" TRS jack, servo balanced
- Input impedance Line: > 5 kOhm
- Input sensitivity through jumper: Lo Gain, +4 dBu, -10 dBV
- Input level for 0 dBFS @ Lo Gain: +19 dBu
- Input level for 0 dBFS @ +4 dBu: +13 dBu
- Input level for 0 dBFS @ -10 dBV: +2 dBV

DA

- Resolution DA: 24 Bit
- Signal to Noise ratio: 108 dB RMS unweighted, 111 dBA (unmuted)
- THD: < - 98 dB, < 0.0013 %
- THD+N: < -91 dB, < 0.002 %
- Crosstalk: > 100 dB
- Maximum output level DA: +19 dBu
- Frequency response DA @ 44.1 kHz, -0.5 dB: 5 Hz – 20.9 kHz
- Frequency response DA @ 96 kHz, -0.5 dB: 5 Hz - 35 kHz
- Output Line: 1/4" TRS jack, servo balanced
- Output impedance Line: 47 Ohm
- Output level through jumper: Hi Gain, +4 dBu, -10 dBV
- Output level at 0 dBFS @ Hi Gain: +19 dBu
- Output level at 0 dBFS @ +4 dBu: +13 dBu
- Output level at 0 dBFS @ -10 dBV: +2 dBV

5.3 Transfer Modes: Resolution / Bits per Sample

ASIO:

- 24 or 32 bit, 4 byte (stereo 8 byte)

This format is compatible with 16-bit and 20-bit. Resolutions below 24-bit are handled by the audio application.

MME:

- 16 bit, 2 byte (stereo 4 byte)
- 20 bit, 3 byte MSB (stereo 6 byte)
- 20 bit, 4 byte MSB (stereo 8 byte)
- 24 bit, 3 byte (stereo 6 byte)
- 24 bit, 4 byte MSB (stereo 8 byte)
- 32 bit, 4 byte (stereo 8 byte)

Channel Interleave operation is not supported. The transfer of Channel Status bits and RME's TMS (Track Marker Support) will be available in a future driver update. Until then DIGICheck's *Channel Status Display* is deactivated.

6. Hardware Installation

6.1 PCI Interface

Before installing the PCI card, please make sure the computer is switched off and the power cable is disconnected from the mains supply. Inserting or removing a PCI card while the computer is in operation can cause irreparable damage to both motherboard and card!

1. Disconnect the power cord and all other cables from the computer.
2. Remove the computer's housing. Further information on how to do this can be obtained from your computer´s instruction manual.
3. Important: Before removing the card from its protective bag, discharge any static in your body by touching the metal chassis of the PC.
4. Insert the PCI card firmly into a free PCI slot, press and fasten the screw.
5. Replace the computer's housing.
6. Reconnect all cables including the power cord.
7. Connect PCI interface and Multiface using the supplied cable (IEEE1394). This is a standard Firewire cable (6-pin).

6.2 CardBus Card

Before inserting the CardBus card make sure the complete HDSP system is ready for operation!

1. Connect the CardBus card with the Multiface using the supplied cable.
2. Insert the CardBus card with the Hammer logo up into a PCMCIA slot.
3. Plug the power jack of the supplied switching power supply into the connector labeled AUX, on the rear of the Multiface.
4. Connect power cord to power supply, plug into AC outlet. The green LED of the power supply and the red LED of the Multiface will light up.
5. Switch on the notebook and boot the operating system.

The small 15-pin connector of the CardBus card is coded. Only the supplied special cable can be plugged in, and only when the metal sleeve is up. Any kind of violence when plugging in and out can cause damage to the CardBus card.

7. Driver Installation

7.1 Windows 98/SE/ME

After the interface has been installed correctly, connected to the Multiface (see 6. Hardware Installation), and the computer has been switched on, Windows will recognize the new hardware component and start its ‘Add New Hardware Wizard’. Insert the RME Driver CD into your CD-ROM drive, and follow further instructions which appear on your computer screen. The driver files are located in the directory **\HDSP_w98** on the RME Driver CD.

Windows will install the Hammerfall DSP System driver, and will register the card in the system as a new audio device. The computer should now be re-booted.

Unfortunately, there are rare cases where the CD-ROM path (i.e. its drive-letter) has to be typed in again during the copy process.

Multiface can be easily configured using the HDSP System’s Settings dialog (see section 9.1)

7.2 Windows 2000/XP

After the interface has been installed correctly, connected to the Multiface (see 6. Hardware Installation), and the computer has been switched on, Windows will recognize the new hardware component and start its ‘Hardware Wizard’. Insert the RME Driver CD into your CD-ROM drive, and follow further instructions which appear on your computer screen. The driver files are located in the directory **\HDSP_w2k** on the RME Driver CD.

Windows will install the Hammerfall DSP System driver, and will register the card in the system as a new audio device. After a reboot the Multiface is ready for use.

Multiface can be configured easily using the HDSP System’s settings dialog (see section 9.1)

In case the warning messages 'Digital signature not found', 'Do not install driver', 'not certified driver' or similar come up: Don't listen to Microsoft, listen to us and continue with the installation.

7.3 Deinstalling the Drivers

A deinstallation of the HDSP's driver files is not necessary – and not supported by Windows anyway. Thanks to full Plug & Play support, the driver files will not be loaded after the hardware has been removed. If desired these files can then be deleted manually.

Unfortunately Windows Plug & Play methods do not cover the additonal autorun entries of TotalMix, the Settings dialog, and the registering of the ASIO driver. Those entries can be removed from the registry through a software deinstallation request. This request can be found (like all deinstallation entries) in *Control Panel, Software*. Click on the entry 'RME Hammerfall DSP Tray Tools', or 'RME Hammerfall DSP'.

7.4 Linux/Unix

An ALSA driver for Linux/Unix is planned to be available soon. Further information on ALSA is available at

http://www.alsa-project.org

8. Operation and Usage

8.1 Connections

The front of the I/O-box Multiface has the MIDI input and output, the analog stereo output of the digital mixer, and several status LEDs:

MIDI State indicates sent or received data for the MIDI port

Input State indicates a valid input signal separately for each input. RME's exclusive *SyncCheck* shows through a blinking LED, which of the input signals is locked, but not in sync to the others. See chapter 9.2, Clock Modes - Synchronisation.

The red **HOST** LED lights up when the power supply or the computer is switched on, thus signalling the presence of operating voltage. At the same time it operates as Error LED, in case the I/O-box wasn't initialised, or the connection to the interface has been interrupted (Error, cable not connected etc.).

Phones is a low impedance line output of highest quality, which can produce a sufficient volume undistorted even when used with headphones.

The back of the Multiface has the 8 analog inputs and outputs, the power supply connector **AUX** (only needed in CardBus operation), and all digital inputs and outputs:

ADAT I/O (TOSLINK), 1 to 3. The ADAT1 I/O can also be used for optical SPDIF, if this mode is selected in the Settings dialog.

SPDIF I/O coaxial (phono)

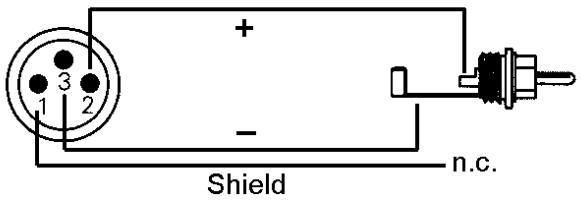
Word clock I/O (BNC)

ADAT Sync In (D-sub 9-pin)

The SPDIF inputs are selected via the Settings dialog (started by clicking on the hammer symbol in the system tray). The HDSP system accepts the commonly used digital audio formats, SPDIF as well as AES/EBU. Channel status and copy protection are ignored.

In SPDIF mode, identical signals are available at both the optical and the coaxial outputs. An obvious use for this would be simply connecting two devices, i.e. using the HDSP as a splitter (distribution 1 on 2).

To receive signals in AES/EBU format, an adapter cable is required. Pins 2 and 3 of a XLR plug are connected individually to the two pins of a phono plug. The cable shielding is only connected to pin 1 of the XLR - not to the phono plug.



The ground-free design using transformers for digital inputs and outputs enables trouble-free connection even to AES/EBU devices, and perfect hum rejection.

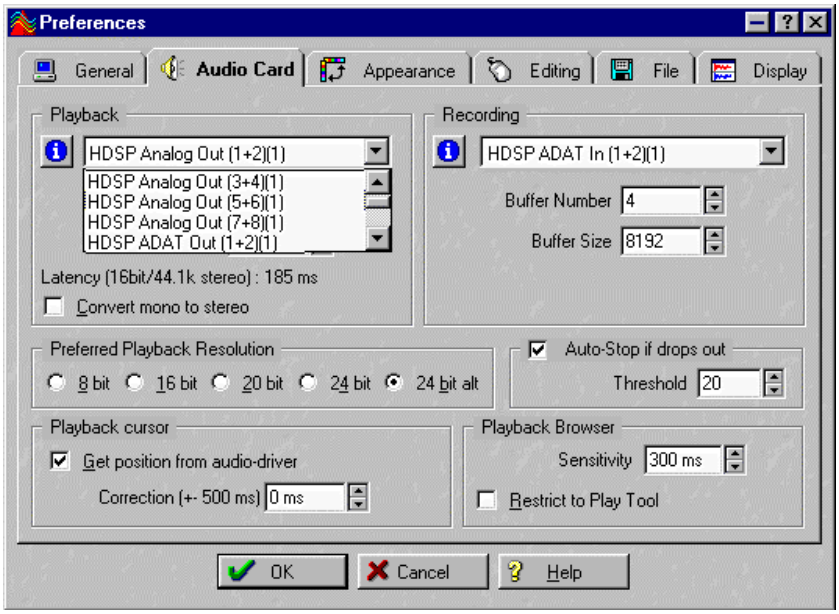
8.2 Playback

The HDSP system can play back audio data only in supported formats (sample rate, bit resolution). Otherwise an error message appears (for example at 22 kHz and 8 bit).

In the audio application being used, HDSP must be selected as output device. This can often be found in the *Options, Preferences* or *Settings* menus under *Playback Device, Audio Devices, Audio* etc. We recommend using 24-bit resolution for playback, to make full use of the HDSP's potential.

We strongly recommend switching all system sounds off (via >Control Panel /Sounds<). Also HDSP should not be the *Preferred Device* for playback, as this could cause loss of synchronization and unwanted noises. If you feel you cannot do without system sounds, you should consider buying a cheap Blaster clone and select this as *Preferred Device* in >Control Panel /Multimedia /Audio<.

The RME Driver CD includes step by step instructions for configuring many popular audio applications, found in the directory **\rmeaudio.web\english\techinfo\conf**.



The screenshot to the right shows a typical configuration dialog as displayed by a (stereo) wave editor. Audio data is sent to either an analog or digital port, depending on which has been selected as playback device.

Increasing the number and/or size of audio buffers may prevent the audio signal from breaking up, but also increases latency i.e. output is delayed.

For synchronized playback of audio and MIDI (or similar), be sure to activate the checkbox 'Get position from audio driver'. Even at higher buffer settings in a mixed Audio/MIDI environment, sync problems will not arise because the Hammerfall DSP always reports the current play position correctly (even while recording - essential for chase lock synchronization).

The HDSP system's ADAT optical interface allows sample rates of up to 96 kHz using a standard ADAT recorder. Single-channel data at this frequency requires two ADAT channels, achieved using the 'Sample Split' technique. This reduces the number of available ADAT channels from 8 to 4. Under Windows MME, channels are routed to ADAT devices in double-speed mode as follows:

- Only stereo pairs (1+2) and (3+4) of the ADAT port are available
- Channel 1 is routed to channels 1 and 2, channel 2 is routed to 3 and 4 etc.

Please refer to the diagram 'ADAT Track Routing, MME 96 kHz', section 23. Routing for record and playback is identical.

8.3 DVD-Playback (AC-3/DTS) under MME

When using popular DVD software player like WinDVD and PowerDVD, their audio data stream can be send to any AC-3/DTS capable receiver, using the HDSP's SPDIF output. For this to work the SPDIF output wave device has to be selected in 'Control Panel/Sounds and Multimedia/Audio'. Also check 'use preferred device only'.

You will notice that the DVD software's audio properties now allow to use 'SPDIF Out' or to 'activate SPDIF output'. When selecting these, the software will transfer the non-decoded digital multichannel data stream using the RME card.

This 'SPDIF' signal sounds like chopped noise at highest level. Therefore check 'Non-audio' in the card's Settings dialog, to prevent most SPDIF receivers from accepting the signal, and to prevent any attached equipment from being damaged.

Setting the card to be used as system playback device is against common sense, as professional cards are not specialized to play back system sounds, and shouldn't be disturbed by system events. To prevent this, be sure to re-assign this setting after usage, or to disable any system sounds (tab Sounds, scheme 'No audio').

Note: The DVD player will be synced backwards from the RME card. This means when using AutoSync and/or word clock, the playback speed and pitch follows the incoming clock signal.

8.4 Low Latency under MME (Buffer Size Adjustment)

Using Windows 95 or 98 the MME buffer size was nothing to worry about. Latencies below 46 ms were not possible. Meanwhile both computers and operating system have become much more powerful, and since Windows ME/2000/XP latencies far lower can be used. SAWStudio and Sonar allowed to use such low settings from the start. Sequoia was updated in version 5.91, WaveLab in version 3.04.

In the HDSP's Settings dialog the MME buffersize (in fact the DMA buffer size) is set with the same buttons as the ASIO buffer size. Our test computers allow to use settings down to 64 samples without clicks. Please note that this setting only defines the buffer size of the hardware. The true and effective latency is configured within the MME application!

Attention: the DMA buffers must not be larger than the application's buffers. This case can happen unnoticed when using ASIO and MME at the same time (multi-client) and setting ASIO to 186 ms, while the buffers in the MME application are still set for a lower latency. Playback will be stuttering and audio will be distorted.

Example: when you set the Hammerfall to 512 you can't use 128 in any program. But setting DMA to 128 allows to use 128 and all higher values within the software.

Please also note that this is a 'you're welcome to try' feature. We can't guarantee that you will be able to use 3 or 6 ms with MME. Simply check out by yourself which lowest setting your system and software allows. Some motherboards with insufficient PCI bandwidth (especially VIA based) suffer from crackling at settings below 512. Be sure to set the buffer size to 512 or higher in such a case.

8.5 Recording Digital

Unlike analog soundcards which produce empty wave files (or noise) when no input signal is present, digital I/O cards always need a valid input signal to start recording.

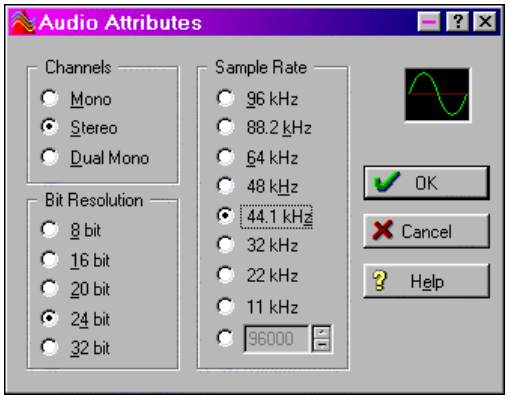
To take this into account, RME has included three unique features in the Hammerfall DSP system: a comprehensive I/O signal status display (showing sample frequency, lock and sync status) in the Settings dialog, status LEDs for each input, and the protective *Check Input* function.

If a 48 kHz signal is fed to the input and the application is set to 44.1 kHz, *Check Input* stops the system from recording. This prevents faulty takes, which often go unnoticed until later on in the production. Such tracks appear to have the wrong playback rate - the audio quality as such is not affected.

The sample frequency shown in the Settings dialog (see chapter 9, screenshot Settings) is useful as a quick display of the current configuration (the board itself and all connected external equipment). If no sample frequency is recognized, it will read 'No Lock'.

With this configuring any suitable audio application for digital recording is simple. After selecting the required input, Hammerfall DSP displays the current sample frequency. This parameter can then be changed in the application's audio attributes (or similar) dialogue.

The screenshot to the right shows a typical dialog used for changing basic parameters such as sample frequency and resolution in an audio application.



Any bit resolution can be selected, providing it is supported by both the audio hardware and the software. Even if the input signal is 24 bit, the application can still be set to record at 16-bit resolution. The lower 8 bits (and therefore any signals about 96dB below maximum level) are lost entirely. On the other hand, there is nothing to gain from recording a 16-bit signal at 24-bit resolution - this would only waste precious space on the hard disk.

It often makes sense to monitor the input signal or send it directly to the output. This can be done at zero latency using **TotalMix** (see chapter 15).

TotalMix also includes a useful automatic real-time monitor function, see chapter 15.8 for details. Activating record in the application causes the input signal to be routed according to the current mixer settings.

Currently two solutions exist which enable an automated control of real-time monitoring. ZLM (Zero Latency Monitoring) allows monitoring in Punch I/O mode - with this the system behaves like a tape machine. This method has been implemented in all versions of Samplitude (by SEK'D), and can be activated using the global track option 'Hardware monitoring during Punch'.

The other solution is Steinberg's ASIO protocol with our ASIO 2.0 drivers and all ASIO 2.0 compatible programs. When 'ASIO Direct Monitoring' has been switched on the input signal is routed in real-time to the output whenever Record is started.

8.6 Recording analog

For recordings via the analog inputs the corresponding record device has to be chosen (HDSP Analog (x+x)). Apart from the internal jumpers which set the basic operating level, the Multiface has no means to change the input level. This would make no sense for the digital inputs, but also for the analog inputs one can do without it. It doesn't matter if the Multiface is operated at a mixing desk or a multichannel Mic preamp, in either case the level can be controlled directly at the source to match the Multiface's sensitivity perfectly.

The input sensitivity of the analog inputs can be changed through internal jumpers to meet the most often used studio levels, see next chapter.

8.7 Analog Inputs

The Multiface provides 8 balanced Line inputs via 1/4" TRS (stereo) jacks. The electronic input stage is built in a servo balanced design which handles monaural and stereo jacks correctly. When used unbalanced it automatically corrects the gain by 6 dB.

When using unbalanced cables with stereo TRS jacks, the 'ring' contact of the cable's jack should be connected to pin 1 (ground). Otherwise noise may occur, caused by the unconnected negative input of the balanced input.

One of the main issues when working with an AD-converter is to maintain the full dynamic range within the best operating level. Therefore the Multiface includes internal jumpers which allow a perfect adaptation for all 8 channels seperately to the three most often used studio levels.

The 'standardized' studio levels do not result in a (often desired) full scale level, but take some additional digital headroom into consideration. The amount of headroom is different in different standards and again differently implemented by different manufacturers. Because of this we decided to define the levels of the Multiface in a most compatible way.

Reference	**0 dBFS @**	**Headroom**
Lo Gain	+19 dBu	15 dB
+4 dBu	+13 dBu	9 dB
-10 dBV	+2 dBV	12 dB

The device ships with +4 dBu as factory default. The according headroom meets the latest EBU recommendations for Broadcast usage. At -10 dBV 12 to 15 dB headroom are common practice, each mixing desk operating at -10 dBV is able to send and receive much higher levels. Lo Gain allows to work with high levels, best suited for professional users who prefer to work balanced and at highest levels.

Information on how to change the jumpers can be found in chapter 9.3.

8.8 Analog Outputs

The 8 short circuit protected, low impedance and servo balanced line outputs are available as (stereo) 1/4" TRS jacks. The electronic output stage is built in a servo balanced design which handles monaural and stereo jacks correctly. When used unbalanced it automatically corrects the gain by 6 dB.

To maintain an optimum level for devices connected to the analog outputs the Multiface includes internal jumpers which allow to change the level of all 8 outputs separately.

As with the analog inputs the analog output levels are defined to maintain a problem-free operation with most other devices. The headroom of the Multiface lies between 9 and 15 dB, according to the chosen reference level:

Reference	**0 dBFS @**	**Headroom**
Hi Gain	+19 dBu	15 dB
+4 dBu	+13 dBu	9 dB
-10 dBV	+2 dBV	12 dB

The device ships with +4 dBu as factory default. The according headroom meets the latest EBU recommendations for Broadcast usage. At -10 dBV 12 to 15 dB headroom are common practice, each mixing desk operating at -10 dBV is able to send and receive much higher levels. Lo Gain allows to work with high levels, best suited for professional users who prefer to work balanced and at highest levels.

Information on how to change the jumpers can be found in chapter 9.3.

9. Configuring the Multiface

9.1 General Information

Configuring the HDSP system is done using its own settings dialog. The panel 'Settings' can be opened in two different ways:

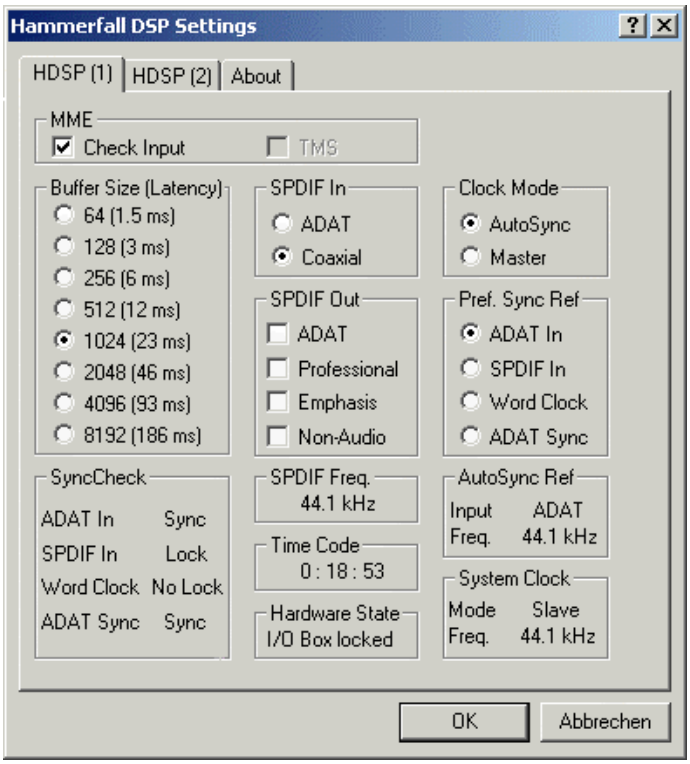
- by clicking on the hammer icon in the Taskbar's system tray
- by starting the 'HDSP_Set' link from the Desktop

The mixer of the Hammerfall DSP System (TotalMix) can be opened in two different ways:

- by clicking on the mixer icon in the Taskbar's system tray
- by starting the 'HDSP_Mix' link from the Desktop

The Hammerfall DSP's hardware offers a number of helpful, well thought-of practical functions and options which affect how the card operates - it can be configured to suit many different requirements. The following is available in the 'Settings' dialog:

- Input selection
- Output mode
- Output channel status
- Synchronization behaviour
- Input and output status display
- Time code display



Any changes made in the Settings dialog are applied immediately - confirmation (e.g. by clicking on OK or exiting the dialog) is not required. However, settings should not be changed during playback or record if it can be avoided, as this can cause unwanted noises. Also, please note that even in 'Stop' mode, several programs keep the recording and playback devices open, which means that any new settings might not be applied immediately.

The status displays at the bottom of the dialog box give the user precise information about the current status of the system, and the status of all signals. 'SyncCheck' indicates whether there is a valid signal for each input ('Lock' or 'No Lock'), or if there is a valid *and* synchronous signal ('Sync'). The 'AutoSync Ref' display shows the input and frequency of the current sync source.

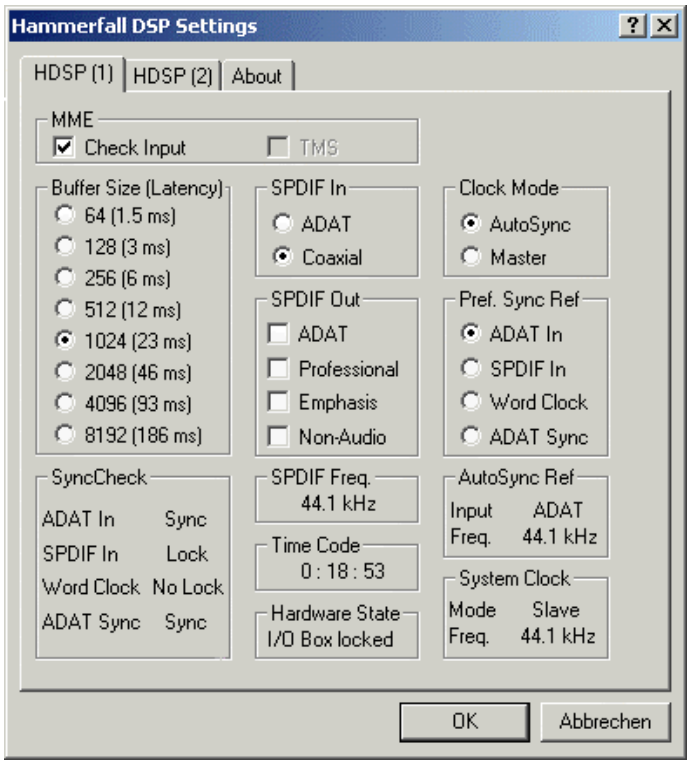
'Time Code' displays time information received from the I/O-box ADAT Sync port. This is convenient for checking whether the system is running in time with the transmitting device (e.g. ADAT).

MME
Check Input verifies the current input signal against the settings in the record program. When de-activated a recording will always be allowed, even with non-valid input signals.
TMS activates the transmission of Channel Status data and Track Marker information of the SPDIF input. Both settings are valid for MME only.

Buffer Size
The setting *Buffer Size* determines the latency between incoming and outgoing ASIO* data, as well as affecting system stability (see chapter 13). Under Windows MME this setting determines the DMA buffer size (see chapter 8.4).

SPDIF In
Defines the input for the SPDIF signal. 'Coaxial' relates to the phono socket, 'ADAT' to the optical TOSLINK input.

SPDIF Out
The SPDIF output signal is constantly available at the phono plug. After selecting 'ADAT' it is also routed to the optical output ADAT. For further details about the settings ‘Professional’, ‘Emphasis’ and ‘Non-Audio’, please refer to chapter 12.



Clock Mode
The card can be configured to use its internal clock (Master), or the clock source pre-defined via *Pref. Sync Ref* (AutoSync).

Pref. Sync Ref
Used to pre-select the desired clock source. If the selected source isn't available the card will change to the next available one. The currently used clock source and sample rate is displayed in the *AutoSyncRef* display.

The automatic clock selection checks and changes between the clock sources ADAT optical, SPDIF, word clock and ADAT Sync. The latter is recommended especially for sample-accurate transfers under ASIO 2.0.

System Clock
Shows the current clock state of the HDSP system. The system is either Master (using its own clock) or Slave (AutoSync Ref).

Hardware State
This display shows the current state of the I/O-box:
I/O Box error: I/O-box not connected or missing power
I/O Box detected: The interface has found a I/O-box and tries to load the firmware
I/O Box locked: Communication between interface and I/O-box ok

* Under W2k/XP also GSIF data

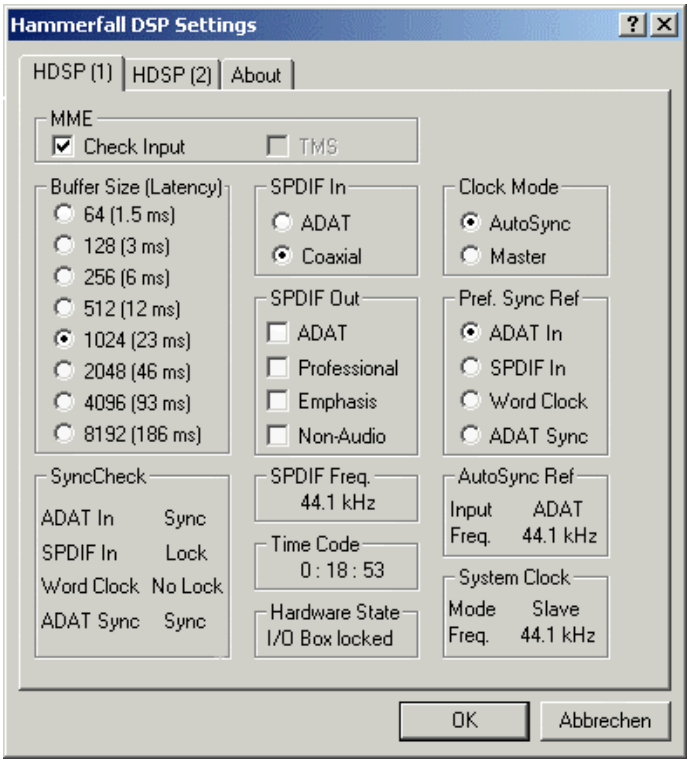
9.2 Clock Modes - Synchronization

In the digital world, all devices are either the ‘Master’ (clock source) or a ‘Slave’ synchronized to the master. Whenever several devices are linked within a system, there must always be a single master clock. The Hammerfall DSP’s intelligent clock control is very user-friendly, being able to switch between clock modes automatically. Selecting 'AutoSync' will activate this mode.

In AutoSync mode, the system constantly scans all digital inputs for a valid signal. If this signal corresponds with the current playback sample rate, the card switches from the internal quartz (AutoSync Ref displays 'Master') to a clock generated from the input signal (AutoSync Ref displays 'Slave'). This allows on-the-fly recording, even during playback, without having to synchronize the card to the input signal first. It also allows immediate playback at any sample rate without having to reconfigure the card.

AutoSync guarantees that normal record and record-while-play will always work correctly. In certain cases however, e.g. when the inputs and outputs of a DAT machine are connected directly to the Hammerfall DSP, AutoSync may cause feedback in the digital carrier, so synchronization breaks down. To remedy this, switch the HDSP’s clock mode over to 'Master'.

Remember that a digital system can only have one master! If the HDSP’s clock mode is set to 'Master', all other devices must be set to ‘Slave’.



All the ADAT optical inputs in the Hammerfall DSP as well as the SPDIF input will work simultaneously. Because there is no input selector however, the HDSP has to be told which of the signals is the sync reference (a digital device can only be clocked from a *single* source). This is why the system has been equipped with automatic clock source selection, which adopts the first available input with a valid digital signal as the clock reference input. The input currently used as sync reference is shown in the *AutoSync Ref* status field, together with its sample frequency.

Via *Pref. Sync Ref* (preferred synchronization reference) a preferred input can be defined. As long as the card sees a valid signal there, this input will be designated as the sync source, otherwise the other inputs will be scanned in turn. If none of the inputs are receiving a valid signal, the card automatically switches clock mode to ‘Master’.

To cope with some situations which may arise in studio practice, setting ‘Pref Sync Ref’ is essential. One example: An ADAT recorder is connected to the ADAT1 input (ADAT1 immediately becomes the sync source) and a CD player is connected to the SPDIF input. Try recording a few samples from the CD and you will be disappointed. Few CD players can be synchronized. The samples will inevitably be corrupted, because the signal from the CD player is read with the (wrong) clock from the ADAT i.e. out of sync. In this case, 'Pref Sync Ref' should be temporarily set to SPDIF.

If several digital devices are to be used simultaneously in a system, they not only have to operate with the same sample frequency but also be synchronous with each other. This is why digital systems always need a single device defined as 'master', which sends the same clock signal to all the other ('slave') devices. RME's exclusive *SyncCheck* technology (first implemented in the Hammerfall) enables an easy to use check and display of the current clock status. The 'SyncCheck' field indicates whether no signal ('No Lock'), a valid signal ('Lock') or a valid *and* synchronous signal ('Sync') is present at each of the digital clock source inputs. The 'AutoSync Ref' display shows the current sync source and the measured frequency.

In practice, SyncCheck provides the user with an easy way of checking whether all digital devices connected to the system are properly configured. With SyncCheck, finally anyone can master this common source of error, previously one of the most complex issues in the digital studio world.

Thanks to its AutoSync technique and lightning fast PLLs, the HDSP is not only capable of handling standard frequencies, but also any sample rate between 25 and 105 kHz. Even the word clock input, which most users will use in varispeed operation, allows any frequency between 25 kHz and 103 kHz.

At 88.2 or 96 kHz: If one of the ADAT inputs has been selected in 'Pref Sync Ref', the sample frequency shown in the 'SPDIF In' field differs from the one shown in 'AutoSync Ref'. The card automatically switches to its Sample Split mode here, because ADAT optical inputs and outputs are only specified up to 48 kHz. Data from/to a single input/output is spread over two channels, the internal frequency stays at 44.1 or 48 kHz. In such cases, the ADAT sample frequency is only half the SPDIF frequency.

9.3 Changing the Jumper Settings

The Multiface has internal jumpers, which allow to change input sensitivity and output level per channel. More information on these settings can be found in chapter 8.5 and 8.6.

Please note that those jumpers are not thought of to be changed every day. They should be changed when the unit is operated for the first time, so that it matches the Studio's operating level. Else only change them if it is really neccessary. The factory default +4 dBu will in most cases offer perfect results.

To change the jumper settings the Multiface must be opened. If you feel unsecure to do so please consult a technician and let him show you how to do it. Else please follow the instructions below step by step.

1. Remove all jacks and cables from the Multiface.
2. Loosen and remove both screws of the ADAT Sync D-sub socket.
3. Use a screwdriver (Phillips 1) to remove the 6 screws on the cover, and the upper middle screw in the front panel of the Multiface, so that the cover can be taken off.
4. Put the device so that the front panel is in front of you. Lift the cover at the front by about one centimeter (0.5 inch). Then pull the cover slowly about 2 centimeter (1 inch) in your direction. When doing so the TRS jacks and the D-Sub socket will slide out of the rear panel. The cover is now freed from the rest of the housing and can be turned to the right.

Use this unique moment and have a look at the internal design of the Multiface. On the right side of the lower printed circuit board (PCB) you can see the two switching power supplies. They generate both 5 Volts (for the digital circuitry) and ±13 Volts (for the analog circuitry) from nearly every possible input voltage. In the center you'll see the heart of the Multiface, the Xilinx FPGA. The analog circuitry and DA-converter for the headphone output is located on the right from the FPGA. Left from the FPGA two Low Jitter PLLs can be seen. On the outer left side you'll find 16 capacitors of the analog outputs and a 50-pin flat cable connector. The flat cable connects lower and upper board, the latter being the analog board, which is mounted down-under at the cover.

The analog board is covered from a thin metalized shield, preventing noise from digital circuitry and flat cable being coupled into analog inputs and outputs. The shield is flexible and can be bended up, so that the view onto the analog board gets free.

But now back to the jumper settings.

6. Bend the flexible shield carefully upwards. You'll now see the analog board with its 16 jumpers for level settings.
7. The jumpers controlling the sensitivity of the inputs are located directly behind each TRS input jack. They allow three different settings: Left (middle plus left pin), right (middle plus right pin), and without jumper. If the device is still placed with the front panel to the front, then Left means +4 dBu (factory default), right means Lo Gain, and without jumper means –10 dBV.
8. The jumpers controlling the output level are located on the other side of the board, and are placed in pairs. The jumper more near to the center of the board is the one of the even channel (2/4/6/8). Again three settings are possible: Left (middle plus left pin), right (middle plus right pin), and without jumper. If the device is still placed with the front panel to the front, then Left means +4 dBu (factory default), right means Lo Gain, and without jumper means –10 dBV.

A drawing showing the jumper position for each level setting is found on the left side (underneath the flat cable) on the analog board.

To prevent the loss of jumpers in –10 dBV mode, we recommend not to remove them completely, but to mount them at the outside of the jumper (no connection to the center pin).

Now you are ready to re-assemble the Multiface.

9. Turn the cover to the left and move it back over the Multiface.

Take care that the flat cable has the same accurate shape as when opening it. Else the flat cable will cause mechanical problems while re-assembling!

10. Move the cover so that it is placed 2 centimeters (1 inch) above the housing. Tilt the cover so that the jacks point to the holes in the rear panel. Carefully slide the jacks into the holes by moving the cover away from you. When inserted completely into the rear panel, the cover can now be layed down.

11. Re-fit the 6 screws into the cover, and re-fit and tighten both screws of the D-sub jack. That's it!

10. Word Clock

10.1 Technical Description and Usage

Correct interpretation of digital audio data is dependent upon a definite sample frequency. Signals can only be correctly processed or transferred between devices if these all share the same clock, otherwise digital signals are misinterpreted, causing distortion, clicks/crackle and even dropouts.

AES/EBU, SPDIF and ADAT are self-clocking, so an additional line for word clock could be considered redundant. In practice however, using several devices at the same time can cause problems. For example, if devices are connected in a loop without there being a defined 'master' device, self-clocking may break down. Besides, the clocks of all devices must be synchronized from a single source. Devices without SPDIF inputs (typically playback devices such as CD players) cannot be synchronized via self-clocking.

In digital studios, synchronization requirements can be met by connecting all devices to a central sync source. For instance, the master device could be a mixing desk, sending a reference signal - word clock - to all other devices. However, this will only work if all the other devices have word clock inputs (e.g. some professional CD players) allowing them to run as slaves. This being the case, all devices will receive the same clock signal, so there is no fundamental reason for sync problems when they are connected together.

10.2 Cables and Termination

Word clock signals are usually distributed in the form of a network, split with BNC T-adapters and terminated with resistors. We recommend using off-the-shelf BNC cables to connect all devices, as this type of cable is used for most computer networks. You will find all the necessary components (T-adapters, terminators, cables) in most electronics and/or computer stores.

To avoid voltage loss and reflections, both the cable itself and the terminating resistor should have an impedance of 75 Ohm. If the voltage is too low, synchronization will fail. High frequency reflection effects can cause both jitter and sync failure.

In practice, the situation has improved in recent years. The relatively low frequency of word clock signals is not a problem for modern electronic circuits. Because of the higher voltage, word clock networks are often more stable and reliable if cables are not terminated at all. Also, 75 Ohm cable is almost impossible to find these days. 50 Ohm cable is standard - this will also work as long as the termination resistors are 75 Ohm.

The word clock input of the Hammerfall DSP is a high-impedance type ensuring maximum flexibility, and is therefore not terminated. If normal termination is necessary (e.g. because Hammerfall DSP is the last device in the chain), simply connect a T-adapter to its BNC input jack, connect the cable supplying the word clock signal to one arm of the T-adapter and terminate the other with a 75 Ohm resistor (as a short BNC plug).

In case Hammerfall DSP resides within a chain of devices receiving word clock, plug a T-adapter into Hammerfall DSP's BNC input jack and the cable supplying the word clock signal to one end of the adapter (as above), but connect the free end to the next device in the chain via a further BNC cable. The last device in the chain should be terminated using another T-adapter and a terminator plug as described in the previous paragraph.

10.3 General Operation

The green ‘Lock’ LED (Input State) will light up when the input sees a valid word clock signal. Selecting ‘Word Clock’ in the ‘Clock Mode’ field will switch clock control over to the word clock signal. As soon as there is a valid signal at the BNC jack, 'AutoSync Ref' will display 'Word'. This message has the same function as the green ‘Lock’ LED, but appears on the monitor, i.e. the user can check immediately whether a valid word clock signal is present and is currently being used.

The word clock output as well as all ADAT ports only works in Single Speed mode. At 96 kHz, the word clock output will therefore be a 48 kHz signal.

11. Using more than one Hammerfall DSP

The current drivers support multiple interfaces and any combination of I/O-boxes. Please note that only one ADAT Sync can be used (of course). Additional all systems must be in sync i.e. have to receive valid sync information (either via wordclock or using AutoSync).

12. Special Characteristics of the SPDIF Output

Apart from the audio data itself, digital signals in SPDIF or AES/EBU format have a header containing channel status information. False channel status is a common cause of malfunction. The Hammerfall DSP ignores the received header and creates a totally new one for the output signal.

Note that in record or monitor modes, set emphasis bits will disappear. Recordings originally done with emphasis should always be played back with the emphasis bit set!

This can be done by selecting the 'Emphasis' switch in the Settings dialogue ('SPDIF Out'). This setting is updated immediately, even during playback. The Hammerfall DSP’s new output header is optimized for largest compatibility with other digital devices:

- 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz or 96 kHz, depending on the current sample rate
- Audio use, Non-Audio
- No Copyright, Copy Permitted
- Format Consumer or Professional
- Category General, Generation not indicated
- 2-channel, No Emphasis or 50/15 µs
- Aux bits Audio Use

Professional AES/EBU equipment can be connected to the Hammerfall DSP thanks to the transformer-balanced coaxial outputs, and the ‘Professional’ format option with doubled output voltage. Output cables should have the same pinout as those used for input (see section 8.1 ‘Connections’), but with a male XLR plug instead of a female one.

Note that most consumer-orientated equipment (with optical or phono SPDIF inputs) will only accept signals in ‘Consumer’ format!

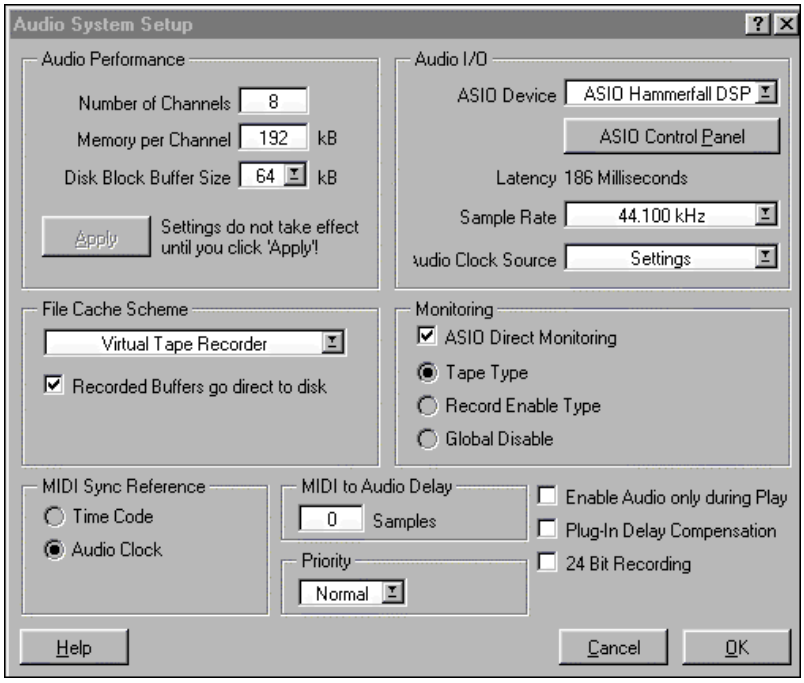
The audio bit in the header can be set to 'Non-Audio'. This is necessary when Dolby AC-3 encoded data is sent to external decoders (surround-sound receivers, television sets etc. with AC-3 digital inputs), as these decoders would otherwise not recognize the data as AC-3.

13. Operation under ASIO 2.0

13.1 General

We will use Steinberg's Cubase VST as an example throughout this chapter. All information provided can easily be adopted to other programs.

Start the ASIO software and select 'System' from the Audio menu. Select 'ASIO Hammerfall DSP' as the audio I/O device. The 'ASIO system control' button opens the HDSP's Settings dialog (see chapter 9, Configuration).



Hammerfall DSP also allows simultaneous record and playback of SPDIF audio data together with record and playback in ADAT format. Please note that the external SPDIF devices have to be running in sync, otherwise recordings will be corrupted.

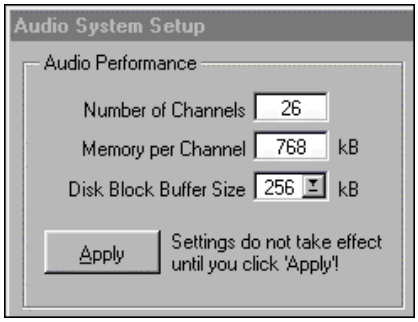
Hammerfall DSP supports 'ASIO Direct Monitoring' (ADM). Please note that at this time Cubase, Nuendo and Logic do not support ADM correctly. Bugfixes should be available soon.

When the sample frequency is set to 88.2 or 96 kHz, the ADAT optical input and output operate in Sample Split mode, so the number of available channels is reduced from 8 to 4.

13.2 Performance

The 'Audio Performance' settings are especially important. Firstly, the number of channels should be changed from 8 to 18 so that all the Hammerfall DSP's inputs can be accessed.

A very common problem is insufficient hard disk performance. If the first track is missing while recording multiple tracks, or the error message 'Audio: Record Error' appears, the disk sub-system is too slow i.e. it is unable to write the audio data to the disk quickly enough. The problem can almost always be remedied by changing 'Disk Block Buffer Size' from the default 64kB to 256kB.


Audio System Setup
Audio Performance
Number of Channels 26
Memory per Channel 768 kB
Disk Block Buffer Size 256 kB
Apply
Settings do not take effect until you click 'Apply'!

This is especially true if you want to record more than 12 tracks at the same time. 26 tracks are only possible after changing 'Disk Block Buffer Size' to 256kB (depending on your computer). Please note that these parameters are only updated after clicking on 'Apply'.

The heyday of (expensive) SCSI hard disks in high-speed audio workstations is over. Today's cheap high-capacity EIDE disks allow continuous transfer rates of well over 10 MB per second. In practical terms, this is more than enough to record up to 24 simultaneous tracks using Cubase and Hammerfall!

However, the hard disks have to work using Busmaster drivers. To activate the EIDE Busmaster mode in Windows 9x, open the Device Manager (Control Panel/System). Double-click on 'Disk drives', then on the required hard disk(s). Select 'DMA' in the Properties dialog, then restart Windows. Under Windows 2000/XP busmaster operation is default.

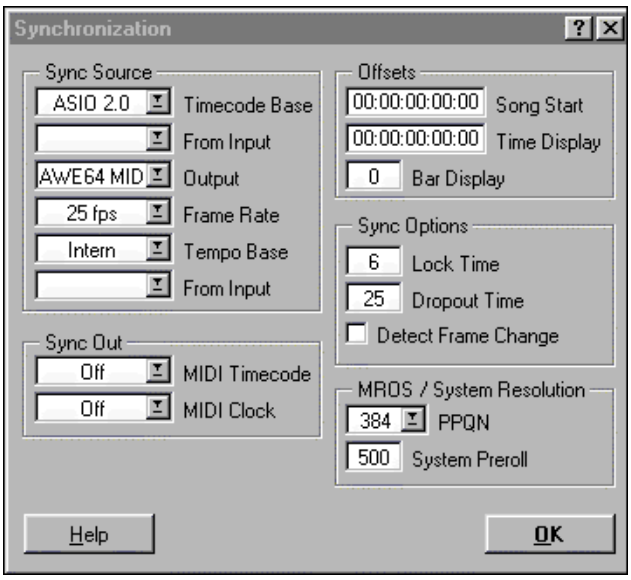
The Buffer Size value in Hammerfall DSP's Settings dialog determines the latency (in this case the delay) between the audio application and the HDSP as well as general system stability. The higher the value, the more tracks can be recorded and played back simultaneously and the longer the system takes to react. At the given maximum of about 0.2 seconds, you will not notice much delay at all - the system will still respond quickly and smoothly.

Most present systems are unable to use the 1.5 ms mode without audible clicks. Current PCs can handle 3 ms. For optimum reliability we recommend setting the highest latency possible, 186 ms.

13.3 Synchronization

To achieve sample-accuracy between the ADAT recorder and Hammerfall DSP while running Cubase, connect the ADAT sync output with the 9-pin D-type sync input of the HDSP. The 'Time Code' field in the Settings dialogue should now show the same position as the ADAT recorder.

Double-clicking on the Sync button in Cubase's transport panel will open the 'Synchronization' dialog. Select ASIO 2.0 as the timecode base (under Sync Source), confirm the dialog with 'OK', then activate Sync mode by (single) clicking on the Sync button.



If synchronization is not working i.e. Cubase does not respond when the ADAT is set to 'Play', please try the following:

- Check the cables
- Switch Sync off and on again (in Cubase's transport panel)
- Select 'Reset Devices' from the Options menu.
- Switch on the ADAT recorder(s) before starting Cubase
- Use the BRC as Master and send its word clock to all other devices
- Use the Clock Mode *ADAT Sync*

13.4 Known Problems

In case the used computer has no sufficient CPU-power and/or sufficient PCI-bus transfer rates, then drop outs, crackling and noise will appear. We also recommend to deactivate all PlugIns to verify that these are not the reason for such effects.

Unfortunately some newer UltraATA66 and UltraATA100 hard disk controller (also Raid controller) seem to violate against the PCI specs. To achieve the highest throughput they hog the PCI bus, even in their default setting. Thus when working with low latencies heavy drop outs (clicks) are heard. Try to solve this problem by changing the default setting of the controller (for example by reducing the 'PCI Bus Utilization').

Another common source of trouble is incorrect synchronization. ASIO does not support asynchronous operation, which means that the input and output signals must not only have the same sample frequency, but they must also be in sync. All devices connected to the Hammerfall DSP must be properly configured for Full Duplex operation. As long as SyncCheck (in the Settings dialog) only displays 'Lock' instead of 'Sync', the devices have not been set up properly!

14. Operation under GSIF (Gigasampler Interface)

14.1 Windows 98/SE/ME

The GSIF interface of the Hammerfall DSP's Windows 98/SE/ME driver allows direct operation with Gigasampler and Gigastudio, with up to 18 channels, 96kHz and 24bit. Additionally the driver supports multi-client operation. For example ASIO can use channels 1/2 and Gigastudio (with GSIF) channels 3/4 simultaneously, and so on.

Gigasampler/Studio requires a lot of the computer's calculation power. An optimum performance is achieved with a stand-alone GSIF PC. If this is not an option we recommend to set the ASIO latency to the highest value (186 ms). This should allow you to achieve a problem-free simultaneous operation of ASIO and GSIF. Gigastudio itself will still work at a very low latency.

Please note: The HDSP system requires identical formats when used in multi-client operation! All programs simultaneously accessing the hardware MUST use the same sample rate.

Simultaneous operation of GSIF and ASIO requires to use different channels. As Cubase VST always uses tracks 1/2 these tracks must not be activated in Gigastudio/Sampler. Additionally the tracks activated in Gigastudio/Sampler have to be de-activated in the ASIO software (for example in the Master Bus section of Cubase).

14.2 Windows 2000/XP

Basically as under Windows 9x. Differences: GSIF under W2k/XP uses a modified interface, which needs interrupts (similar to ASIO). Therefore the user can now set and change the latency (under W9x latency was fixed inside Gigastudio). However, when using the Hammerfall DSP, the latency is always the same as the one selected for ASIO operation. This can cause performance problems on slower machines when using GSIF and ASIO at the same time.

Please note that the W2k/XP driver fully supports multiclient operation, including the combination MME/ASIO. So for example Cubase, Gigastudio and Sonar can be used simultaneously, provided each of these programs uses its own audio channels exclusively. Please also note that Gigastudio is running unexpectedly in the background (thus blocking its assigned audio channels), as soon as the Gigastudio MIDI ports are used – even when Gigastudio itself hasn't been started.

15. TotalMix: Routing and Monitoring

The Hammerfall DSP system includes a powerful digital real-time mixer. RME's unique TotalMix technology allows for nearly unlimited mixing and routing with all inputs and playback channels simultaneously.

Here are some typical applications for TotalMix:

- setting up delay-free submixes (headphone mixes)
- unlimited routing of inputs and outputs (free utilisation, patchbay function)
- distributing signals to several outputs at a time
- simultaneous playback of different programs over only one stereo channel
- mixing of the input signal to the playback signal (complete ASIO Direct Monitoring)
- integration of external devices (effects etc). in real-time
- mixdown of three ADAT inputs to one (realizing two additional inputs)

On page 42 you'll find a block diagram of the TotalMix mixer of the Multiface. It can help to understand the basic signal flow and routing. It shows that the record signal always stays unaltered, but can be passed on as often as desired, even with different levels. The level meter of inputs and playback channels are connected pre-fader (due to the enormous routing capabilities). The level meters of the hardware's outputs are connected post-fader.

15.1 Elements of the Surface

The visible design of the mixer is mainly determined by the architecture of the HDSP system:

- Upper row: hardware inputs. The level shown is that of the input signal, i. e. Fader independent. Per fader and routing window, any input channel can be routed and mixed to any hardware output (third row).
- Middle row: playback channels (playback tracks of the software). Per fader and routing window, any playback channel can be routed and mixed to any hardware output (third row).
- Lower row: hardware outputs. Because they refer to the output of a subgroup, the level can only be attenuated here (in order to avoid overloads), routing is not possible. This row has two additional channels, the analog outputs.

Every single channel has various elements:

Input and playback channels each have a mute and solo button.

Below each there is the panpot, realized as indicator bar (L/R) in order to save space.

In the window below this, the present level is displayed in RMS or Peak, being updated about every half a second. Overs are indicated here by an additional red dot.

Then comes the fader with a levelmeter. The meter shows both peak values (zero attack, 1 sample is enough for displaying full scale) by means of a yellow line and mathematically correct RMS values by means of a green bar. The RMS display has a relatively slow time constant, so that it shows the average loudness quite well.

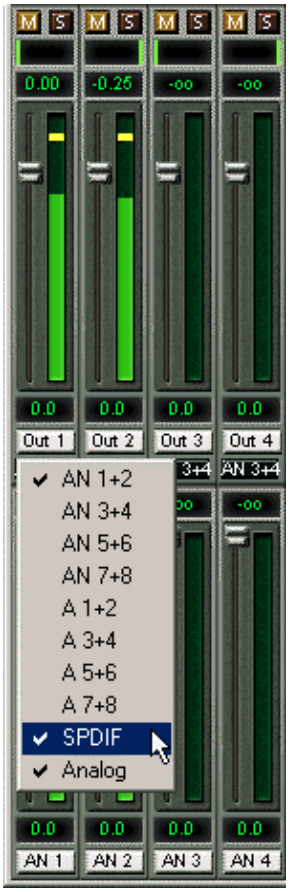
Below the fader, the current gain and panorama values are shown.

The white area shows the channel name, the black area shows the current routing target.

15.2 Tour de TotalMix

In the following chapters we will explain all functions of the surface step by step. Starting up TotalMix, the last settings are recalled automatically. When executing the application for the first time, a default file is loaded, sending all playback tracks 1:1 to the corresponding hardware outputs with 0 dB gain. The faders in the upper row are set to maximum attenuation (called m.a. in the following), so there is no monitoring of the input channels.

We will now create a small submix for the analog headphone output. Please start a multitrack playback and connect your headphones to the headphone output. In playback channel 1 (labeled 'Out 1'), click onto the routing window below the label. A list pops up, showing a checkmark in front of 'AN 1+2'. Click onto 'Analog'. The list disappears, the routing window no longer shows 'AN 1+2', but 'Analog'. Now move the fader with the mouse. As soon as the fader value is unequal m.a., the present state is being stored and routing is activated. Move the fader button to around 0 dB. The present gain value is displayed below the fader in green letters. In the lower row, on channels 27 and 28 (AN.L. and AN.R.), you can also see the level of what you are hearing in the phones. The level meter of the hardware output shows the outgoing level. Click into the area above the fader and drag the mouse in order to set the panorama, in this case the routing between channels 27 and 28. The present pan value is also being displayed below the fader.



Please carry out the same steps for 'Out 2' now, in order to route it to the headphone output as well.

Often signals are stereo, i. e. a pair of two channels. It is therefore helpful to be able to make the routing settings for two channels at once. Press the Ctrl-key and click into the routing window of 'Out 3' with the key pressed. The routing list pops up with a checkmark at 'AN 3+4'. Click onto 'Analog'. Now, channel 4 has already been set to 'Analog' as well.

When you want to set the fader to exactly 0 dB, this can be difficult, depending on the mouse configuration. Move the fader close to the 0 position and now press the Shift-key. This activates the fine-mode, which stretches the mouse movements by a factor of 8. In this mode, a gain setting accurate to 0.1 dB is no problem at all.

Please set 'Out 4' to a gain of around -20 dB and the pan close to center. Now click onto the routing window. You'll now see two checkmarks, one at 'AN 3+4', the other one at 'Analog'. Click onto 'SPDIF'. The window disappears, fader and panpot jump to their initial values, the signal can now be routed to the SPDIF output. You can continue, until all entries have got a checkmark, i. e. you can send the signal to all outputs simultaneously. This is one of several differences to the Cubase mixer, which does not allow for multiple selections.

You will certainly have noticed that the headphone mix has not changed, while you were routing the channel to other outputs and setting different gain values. With all analogue and most digital mixing desks, the fader setting would affect the level for every routed bus - not so for TotalMix. TotalMix allows for setting all fader values individually. Therefore the faders and the panpots jump to the appropriate setting as soon as another routing is chosen.

The checkmarks are un-checked by moving the fader to m.a. This setting deactivates the routing...why route if there is no level? Click onto 'AN 3+4' in the routing window, pull the fader down, open the routing window again - the checkmark is gone.

15.3 Submix View

Such a wide range of possibilities make it difficult to maintain the overview. Because practically all hardware outputs can be used for different submixes, as shown. And when opening the routing windows you might see an army of checkmarks, but you don't get an overwiev, i.e., how the signals come together and where. This problem is removed by the view mode 'Submix'. In this mode, all routing windows jump to the routing pair just being selected. So you can then see immediately, which channels, which fader and pan settings make a submix (for example 'Analog').

At the same time the Submix View simplifies setting up the mixer, as all channels can be set simultaneously to the same routing destination with just one click.

15.4 Mute and Solo

Mute works pre-fader, thus mutes all active routings of the channel. As soon as any Mute button is pressed, the Master Mute button lights up in the quick access area. It can switch all selected mutes off and on again. You can comfortably make mute groups to activate and deactivate this way.

The same holds true for the Solo and the Master Solo buttons. Solo is working as a solo-in-place. As soon as one Solo button is pressed, all other Mute buttons are activated and light up. But TotalMix would not be an *Intelligent Audio Solution*, if it didn't behave as you'd expect from a mixing console. If you, for instance, mute 'Out 1' to 'Out 4' and press Solo for 'Out 5', of course all Mute buttons will light up. If you deactivate Solo, the Mute buttons for 'Out 1' to 'Out 4' light up as before. And if you chose Solo for a channel of this Mute group, mute will be deactivated, but immediately activated again, if Solo is released.

15.5 Hotkeys

TotalMix knows only a few, but very effective key combinations, that make setting the mixer up considerably easier and faster. The Shift-key for the fine-mode for faders and panpots has already been mentioned. But the Ctrl-key can do far more than changing the routing pairwise:

- Clicking anywhere into the fader area with the Ctrl-key pressed, sets the fader to 0 dB, -6 dB for the hardware outputs.
- Clicking anywhere into the pan area with the Ctrl-key pressed, sets the panorama to <C> meaning 'Center'.

The faders can also be moved pairwise, corresponding to the stereo-routing settings. This can be achieved by pressing the Alt-key and is especially comfortable when setting the SPDIF and analogue output level. Even the Panoramas can be operated with Alt, from stereo through mono to inversed channels. But also the Mute and Solo buttons (ganged or inversed switching!). At the same time, TotalMix also supports combinations of these keys. If you press Ctrl and Alt at the same time, clicking with the mouse makes the faders jump to 0 dB pairwise, and they can be set pairwise by Shift-Alt in fine-mode.

Also very useful: the faders have two mouse areas. The first area is the fader button, which can be grabbed at any place without changing the position. This avoids unwanted changes when clicking onto it. The second area is the whole fader setting area. Clicking into this area makes the fader jump to the mouse at once. If you want to set several faders to m.a. for instance, it is sufficient to click onto the lower end of the fader path. Which happens pairwise with the Alt-key pressed.

Using the hotkeys I, O and P the complete row each of Input, Playback and Output channels can be toggled between visible and invisible. Hotkey S switches Submix view on/off. Those four hotkeys have the same functionality as the buttons in the View section of the Quick Access Panel. The Level Meter Setup dialog can be opened via F2 (as in DIGICheck and the Meter Bridge).

Further hotkeys are available to control the configuration of the Level Meter (see chapter 15.9):

Key 4 or 6: Display range 40 or 60 dB
Key E or R: Numerical display showing Peak or RMS
Key 0 or 3: RMS display absolute or relative to 0 dBFS

15.6 The Quick Access Panel

This section includes additional options, further improving the handling of TotalMix. The Master button for Mute and Solo has already been described, they allow for group-based working with these functions.

In the **View** section the single rows can be made visible or invisible. If the inputs are not needed for a pristine playback mix, the whole upper row falls out of the picture after a click on the input button. If the hardware outputs don't interest you either, the surface can thus be reduced to the playback channels to save space. All combinations are possible.

Submix sets all routing windows to the same selection as described before. Deactivating Submix automatically recalls the previous view.

The mixer can also be made smaller horizontally, and, scrolled. TotalMix can be made substantially smaller and space-saving on the desktop/screen, if you have to have to monitor or set only a few channels or level meters.

The **Presets** are one of the mightiest and most useful features of TotalMix. Behind the eight buttons, eight files are hidden (see next chapter). These contain the complete mixer state. Just try it: all faders and other settings follow the changing of preset(s) in real-time, just by a single mouse click. The Save button allows for storing the present settings in the present preset. You can change back and forth between a signal distribution, complete input monitoring, a stereo and mono mix, and various submixes without any problem.

Also here, RME's love for details can be seen. If any parameter is being altered after loading a preset (e. g. moving a fader), the preset display flashes in order to announce that something was changed, still showing, which state the present mix is based on.

If no preset button is lit, another preset had been loaded via the File menu and 'Open file'. Mixer settings can of course be saved the usual way, and with long file names.

Up to three Hammerfall DSP systems can be used simultaneously. The **Card** buttons switch between the systems. Systems, because card 1 can be a Multiface, but card 2 can also be a Digiface.

The number of ADAT channels is reduced to half automatically when chosing double speed operation (88.2 or 96 kHz). The display is adjusted accordingly, and all fader settings remain stored (even the invisible ones).

15.7 Presets

During the driver installation 8 factory presets will be copied to the directory \Windows. The files are named **preset1.mix** to **preset8.mix**, and will be used when clicking on the 8 Preset buttons in the Quick Access Panel.

But TotalMix will read those files only at first usage. As soon as one of the Presets is saved, TotalMix writes a new file and adds the number of the currently used system (Card 1, 2 or 3). The files **preset1.mix** thus changes to **preset11.mix**, if Card 1 was active. This method offers two major advantages:

- Presets modified by the user will not be overwritten when reinstalling or updating the driver
- The factory presets remain unchanged, and can be reloaded anytime using the menu, Files/Open

The 8 factory presets offer not only a useful functionality for TotalMix, but also a pretty good base to modify them to your personal needs.

Preset1.mix
Description: All channels routed 1:1, playback monitoring via headphone out

Details: All inputs maximum attenuation (m.a.). All playback channels 0 dB, routet to the same output. All output channels 0 dB, phones -6 dB. Submix of all inputs and outputs to the analog output (Phones), with input faders set to m.a., playback to 0 dB. All channels prepared for all routings to left/right panning. Level display set to RMS -3 dB.

Note: This preset is *Default*, offering the standard functionality of a I/O-card.

Preset2.mix
Description: All channels routed 1:1, input and playback monitoring via Phones. As Preset 1, plus submix of all inputs (0 dB) on Phones.

Preset3.mix
Description: All channels 1:1, input and playback monitoring via Phones and outputs. As Preset 2, but all inputs set to 0 dB (1:1 pass through).

Preset4.mix
Description: All channels 1:1, playback monitoring via Phones and outputs. As Preset 3, but all inputs muted.

Preset5.mix
Description: All faders m.a. As Preset 1, but all outputs m.a.

Preset6.mix
Description: Submix on SPDIF at -6 dB. As Preset 1, plus submix of all playbacks on SPDIF. View Submix SPDIF active.

Preset7.mix
Description: Submix on SPDIF at -6 dB. As Preset 6, but submix of all inputs and outputs on SPDIF. View Submix SPDIF active.

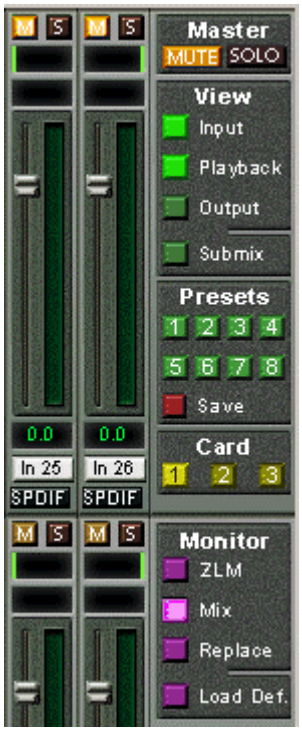
Preset8.mix
Description: Panic. As Preset 4, but also playback muted (no output signal)

15.8 Monitor

The Monitor section of the Quick Access Panel is only valid for our Windows MME driver, i.e. when using programs like WaveLab, Soundforge, Sonar or Samplitude.

Monitor offers two advanced automated monitoring solutions. Monitoring will be controlled either by special commands directly from the recording software (Samplitude/Sequoia/SAWStudio, mode ZLM), or by the recording state itself (mode Mix/Replace).

The basic method used is as simple as it is clever: ZLM and record are controlling the Mute buttons of the corresponding channels. For this to work, Mute must be activated on the record's Input channel. The fader must not be set to m.a..



A click on **Load Def.** will load a template (preset4.mix), which can be used to verify and test this functionality. But it is also possible to use any other mixer state, as long as the recording channels are muted manually.

As soon as a recording starts, the corresponding channels will be unmuted, i.e. the input signal will be processed according to the current mixer settings. TotalMix lets you check the whole process visually, as the Mute buttons in TotalMix will be switched on and off automatically.

In **Mix** mode the input signal will be mixed on the outputs when record is active. In **Replace** mode the Mute button of the corresponding playback channel will be activated, so that the input signal replaces the playback signal.

ZLM is a special function for tape machine style monitoring when doing punch-ins and outs. For this to work the option 'Hardware monitoring during punch' has to be activated in Samplitude/Sequoia. Then at each punch-in the corresponding Mute buttons will be deactivated, at punch-out they will be re-activated.

All settings can be changed and configured in real-time.

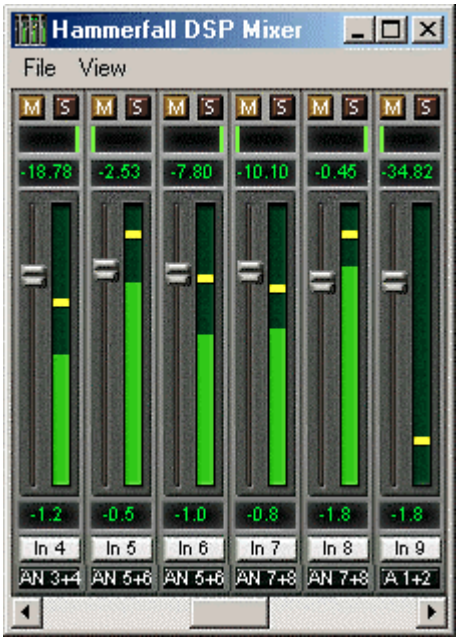
15.9 Level Meter

Having set a new standard with the level meters of DIGICheck, Hammerfall DSP goes even further: The calculation of the Peak, RMS and Over is realized in hardware, in order to be capable of using them independent of the software in use, and to significantly reduce the CPU load.

The level meters integrated in TotalMix - considering their size - cannot be compared with the *HDSP Meter Bridge* (chapter 18.2). Nevertheless they already include many useful functions.

Peak and RMS is displayed for every channel. 'Level Meter Setup' (Menu Options or F2) or direct keyboard entry (*hotkeys*) makes various options available:

- Display range 40 or 60 dB (*hotkey 4 or 6*)
- Release time of the Peak display (Fast/Medium/Slow)
- Numerical display selectable either Peak or RMS (*Hotkey E or R*)
- Number of consecutive samples for Overload display (1 to 15)
- RMS display absolute or relative to 0 dBFS (*Hotkey 3 or 0*)



The latter is a point often overlooked, but nonetheless important. RMS shows 3 dB less for sine signals. This is mathematically correct, but not very reasonable for a level meter. Therefore, we had corrected DIGICheck's RMS display by 3 dB, a full scale sine signal shows both 0 dBFS Peak and RMS. This setting also yields directly readable signal-to-noise values, while other applications (like WaveLab) will show a value 3 dB better than actual (because the reference is not 0 dB, but -3 dB).

The value displayed in the text field is independent of the setting 40/60 dB, it represents the full 24 bit range of the RMS measurement, thus making possible a SNR measurement 'RMS unweighted', which you would otherwise need extremely expensive measurement devices for. An ADI-8 DS connected to the Multiface will therefore show around -113 dB on all 8 channels.

This level display will constantly bring the reduced dynamic range of your equipment, maybe of the whole studio, in front of your eyes. Nice to have everything 24 bit - but still noise and hum everywhere in the range around -90 dB or worse... sorry, but this is hard reality. The up-side about it is that TotalMix allows for constantly monitoring the signal quality without effort. Thus it can be a valuable tool for sound optimization and error removal in the studio.

Measuring SNR (Signal to Noise) requires to press R (for RMS) and 0 (for referring to 0 dBFS, a full scale signal). The text display will then show the same value as an expensive measurement system, when measuring 'RMS unweighted'.

Note: There is no RMS calculation for the third row, the physical outputs. Therefore these green bars show the peak value.

16. Notes on using Laptops and CardBus

The HDSP system uses the notebook's PCMCIA type II port as *CardBus* interface. Compared to a PC-Card, which only has access to the outdated ISA-bus, CardBus is a 32 bit PCI interface. Like with a desktop system it's not possible to remove a PCI device while in operation. First the operating system has to receive a ‚removal request', then the device has to be stopped. Finally the card can be pulled out of the PCMCIA slot.

To remove the hardware click on the green arrow symbol in the systray. It is possible to stop the HDSP directly, or to first call up the info dialog by double clicking the symbol, and then stopping it.

When inserting the CardBus card it usually will be detected automatically by the notebook hardware and then Windows. A double beep signals detection and removal. In rare cases detection will fail. If so, simply remove the card and insert it again. On one of the tested notebooks the card was never activated at Windows boot, but only when plugged in manually. The problem vanished after a fresh install of Windows ME.

The Hammerfall DSP System was tested thoroughly on several notebooks by RME. Basically our CardBus card operates with any notebook we have tested. But there were some significant performance differences, and some laptops suffered from a very low data transfer rate of the PC-Card controller. Detailed information on this can be found in the Tech Info *HDSP System: Notebook Tests – Compatibility and Performance*.

The mobile operation of the HDSP system can cause problems. Explanations and solutions on digital noise, ground loops, headphone operation and Line Out wiring, power supplies and the mobile operation with battery can be found in the Tech Info *HDSP System: Notebook Basics - The Audio Notebook in Practise*.

The hardware of a notebook differs in many points from that of a desktop computer – sometimes…Detailed information on all components, from CPU to the display, can be found in the Tech Info *HDSP System: Notebook Basics – Notebook Hardware*.

Why does your notebook behave completely different than your desktop? And how you can still change it for good performance (not only audio performance)? Find answers to these and many other questions about software tuning and installation, clicks under Win9x, the usage of the low latency solution Windows 2000/XP, and the optimization of the memory bandwidth, in the Tech Info *HDSP System: Notebook Basics – Background Knowledge and Tuning*.

17. Hotline - Troubleshooting

17.1 General

The newest information can always be found on our website www.rme-audio.com, section FAQ, Latest Additions.

The ADAT timecode is not in sync

- The tape is formatted to 48 kHz, but played back at 44.1 kHz (Pitch). This 'Blackface' problem cannot be solved in a satisfactory way.

ADAT timecode is running, but Cubase does not start 'Play' automatically

- The input displayed in ‘Sync Ref’ is not in sync mode. Sync mode is essential, because ADAT’s so-called time code is really a sample position, and is therefore only valid for synchronous audio data.
- Sync is displayed (referring to the card’s clock), but the incoming data is not in sync with the sample position received at the ADAT Sync In. Then Cubase does not start. Remedy: Set ‘Pref. Sync Ref’ to the input corresponding to the received ADAT Sync signal.
- Sync mode wasn't activated (button in the transport panel), or ASIO 2.0 has not been chosen as the SMPTE sync source.

The input signal cannot be monitored in real-time

- ASIO Direct Monitoring has not been enabled, and/or monitoring has been globally disabled.

The 8 ADAT channels do not appear at the output

- The optical output has been switched to SPDIF. Nevertheless you can use the 8 corresponding playback tracks, by using TotalMix and mixing/routing them to other physical outputs.

Playback works, but record doesn’t:

- Check that there is a valid signal at the input. If so, the current sample frequency is displayed in the Settings dialog.
- Check whether the Hammerfall DSP has been selected as recording device in the audio application.
- Check whether the sample frequency set in the audio application (‘Recording properties’ or similar) matches the input signal.
- Check that cables/devices have not been connected in a closed loop. If so, set the systems’s clock mode to ‘Master’.

Crackle during record or playback:

- Increase the number and size of buffers in the ‘Settings’ dialog or in the application.
- Try different cables (coaxial or optical) to rule out any defects here.
- Check that cables/devices have not been connected in a closed loop. If so, set the system’s clock mode to ‘Master’.
- Increase the buffer size of the hard disk cache.
- Activate Busmaster mode for the hard disks (see section 13.2 'Performance').
- In case of a recently done BIOS update of the motherboard: Propably 'Load BIOS Defaults' was loaded instead of 'Load Setup Defaults'. This sets the 'PCI Latency Timer' to 0 (default: 32).

Low Latency ASIO operation under Windows 2000/XP on single CPU systems:

- To use ASIO at lowest latencies under Windows 2000/XP even when only having one CPU, the system performance has to be optimized for background tasks. Go to Control Panel/System/Advanced/Performance Options. Change the default 'Applications' to 'Background tasks'. The lowest usable latency will drop from 23 ms to around 3 ms. This is no issue when using dual CPU systems.

17.2 Installation

More information on installation problems (which fortunately are very seldom, thanks to Plug and Play), can be found in the Tech Info 'Installation problems'. It is located in the directory **\rmeaudio.web\techinfo** on the RME Driver CD.

Hammerfall DSP is normally found in the Device Manager (>Settings/Control Panel/System<), category 'Sound-, Video- and Gamecontroller'. A double click on ' Hammerfall DSP ' starts the properties dialog. Choosing 'Resources' shows Interrupt and Memory Range.

The newest information on hardware problems can always be found on our website www.rme-audio.com, section FAQ, Hardware Alert: about incompatible hardware.

The dialog 'New hardware component found' does not appear:

- Check whether the CardBus card is completely inserted into the PCMCIA slot, or the PCI interface is correctly inserted in the PCI slot.

The card and drivers have been installed correctly, but playback does not work:

- Check whether the Hammerfall DSP appears in the Device Manager. If the ' Hammerfall DSP' device has a yellow exclamation mark, then there is an address or interrupt conflict.
- Even if there is no yellow exclamation mark, it is worth checking the 'Resources' tab anyway.
- Check whether the Hammerfall DSP has been selected as current ASIO device.

The computer crashes whenever the Hammerfall DSP is accessed:

- If your graphics board is an older Matrox Mystique or uses a '968' S3 chip, there could be a memory allocation error. Change the memory area allocated to the Hammerfall DSP via /Resources/Change Setting. Detailed information on this subject can be found in **\rmeaudio.web\techinfo\install.htm** on the RME Driver CD.

18. HDSP Software

18.1 DIGICheck

The DIGICheck software is a unique utility developed for testing, measuring and analysing digital audio streams. Although the DIGICheck software is fairly self-explanatory, it still includes a comprehensive online help. A detailed description of all functions is also available in HTML format (digich.htm, in the **\techinfo** directory on the RME Driver CD or from our website). The following is a short summary of the available functions:

- **Level Meter**. High precision 24-bit resolution, 2/8/26 channels. Application examples: Peak level measurement, RMS level measurement, over-detection, phase correlation measurement, dynamic range and signal-to-noise ratios, RMS to peak difference (loudness), long term peak measurement, input check.
- **Bit Statistics**. Shows the true resolution of audio signals as well as errors and DC offset.
- **Performance Test**. Measures overall system performance (computer + HDSP)
- **Memory Test**. Tests the Hammerfall hardware and the entire audio data path in the PC

To install DIGICheck, go to the **\DIGICheck** directory on the RME Driver CD and run *setup.exe*. Follow the instructions prompted on the screen.

18.2 HDSP Meter Bridge

The Hammerfall DSP Meter Bridge is a unique, highly flexible and handy tool for level metering. As opposed to DIGICheck, the HDSP Meter Bridge receives all level data directly from the hardware. Thus the Meter Bridge can run in parallel to any other program. As Peak and RMS calculations are performed directly in hardware, the CPU load caused is limited to the graphics routines – and is near zero on todays computers.

Although the HDSP Meter Bridge is fairly self-explanatory, it still includes a comprehensive online help. To start press F2 and F7, the most important hotkeys.

To install the HDSP Meter Bridge, go to the **\HDSP Meter Bridge** directory on the RME Driver CD and run *setup.exe*. Follow the instructions prompted on the screen.

19. Accessories

RME offers several optional components, further increasing the flexibility and usability of the HDSP system. Additionally parts of the HDSP system, like the special CardBus cable and the switching power supply, are available seperately.

Part Number	**Description**
36000	19“, 1UH Universal rack holder

This 19" rack holder has holes for Digiface and Multiface. Two units can be installed side by side in any combination. The rack holder also includes holes for nearly all 19" half-rack units from other manufacturers.

36001	Firewire cable IEE1394 6M/6M, 1 m (3.3 ft)
36002	Firewire cable IEE1394 6M/6M, 3 m (9.9 ft)
36005	Firewire cable IEE1394 6M/6M, 5 m (16.4 ft)
36010	Firewire cable IEE1394 6M/6M, 10 m (32.8 ft)

Firewire cable for the HDSP system, both sides 6-pin male. Cable longer than 16 ft is not allowed for Firewire, therfore hard to get in computer shops. However the HDSP system can operates flawlessly even with a cable length of up to 50ft (15 m).

36081	RME Firewire cable for CardBus 15/6M, 5 m (16.4 ft)
36082	RME Firewire cable for CardBus 15/6M, 1 m (3.3 ft)

Special cable 15-pin close Lan coded to 6-pin male, for RME CardBus card.

36003	Optical cable, Toslink, 0.5 m (1.5 ft)
36004	Optical cable, Toslink, 1 m (3.3 ft)
36006	Optical cable, Toslink, 2 m (6.6 ft)
36007	Optical cable, Toslink, 3 m (9.9 ft)
36008	Optical cable, Toslink, 5 m (16.4 ft)
36009	Optical cable, Toslink, 10 m (32.8 ft)

Standard lightpipe with TOSLINK connectors, RME approved quality.

36011	RME FW Repeater for Digiface/Multiface 6F/6F

Active receiver/transmitter to extend the cable length of the HDSP system. Can not be used as Firewire repeater! Using two Repeaters and 10 m Firewire cables, up to 30 m (100 ft) can be realized between interface and I/O box. Switchable ground lift is also available, to avoid ground loops and disturbances due to different power sources.

37011	Power supply for HDSP CardBus card

Robust and light weigth switching power supply, 100V-240V AC, 12V 1.25 A DC. Also neccessary when operating the Repeater in ground lift mode, cause Digiface and Multiface will no longer be powered from the computer (the PCI interface).

20. TECH INFO

Not all information to and around our products fit in a manual. Therefore RME offers a lot more and detailed information in the **Tech Infos**. The very latest Tech Infos can be found on our website, section News & Infos, or the directory **\rmeaudio.web\techinfo** on the RME Driver CD. These are some of the currently available *Tech Infos*:

Synchronization II (DIGI96 series)
Digital audio synchronization - technical background and pitfalls.

Installation problems
Problem descriptions and solutions.

Information on driver updates
Lists all changes in the drivers.

Configuring Logic, Samplitude, Cubase, Cakewalk, Sonar and SAWPlus32
Step by step instructions for use with RME cards.

DIGICheck: Analysis, tests and measurements with the DIGI96 series
A description of DIGICheck, including technical basics.

ADI-8 Inside
Technical information about the RME ADI-8 (24-bit AD/DA converter).

HDSP System: Notebook Basics - Notebook Hardware
HDSP System: Notebook Basics - The Audio Notebook in Practice
HDSP System: Notebook Basics - Background Knowledge and Tuning
HDSP System: Notebook Tests - Compatibility and Performance
Many background information on and tests of notebooks

HDSP System: TotalMix - Hardware and Technology
HDSP System: TotalMix - Software, features, operation
The digital mixer of the Hammerfall DSP in theory and practise

21. Warranty

Each individual Hammerfall DSP undergoes comprehensive quality control and a complete test in a PC environment at RME before shipping. This may cause very slight signs of wear (if it looks like it was used one time before - it was). The usage of high grade components allows us to offer a full two year warranty. We accept a copy of the sales receipt as valid warranty legitimation.

RME's replacement service within this period is handled by the retailer. If you suspect that your card is faulty, please contact your local retailer. The warranty does not cover damage caused by improper installation or maltreatment - replacement or repair in such cases can only be carried out at the owner's expense.

RME does not accept claims for damages of any kind, especially consequential damage. Liability is limited to the value of the Hammerfall DSP. The general terms of business drawn up by Synthax OHG apply at all times.

22. Appendix

RME news, driver updates and further product information are available on our website:

http://www.rme-audio.com

If you prefer to read the information off-line, you can load a complete copy of the RME website from the RME Driver CD (in the **\rmeaudio.web** directory) into your browser.

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Current driver version: W98: 2.13, W2k/XP: 2.33

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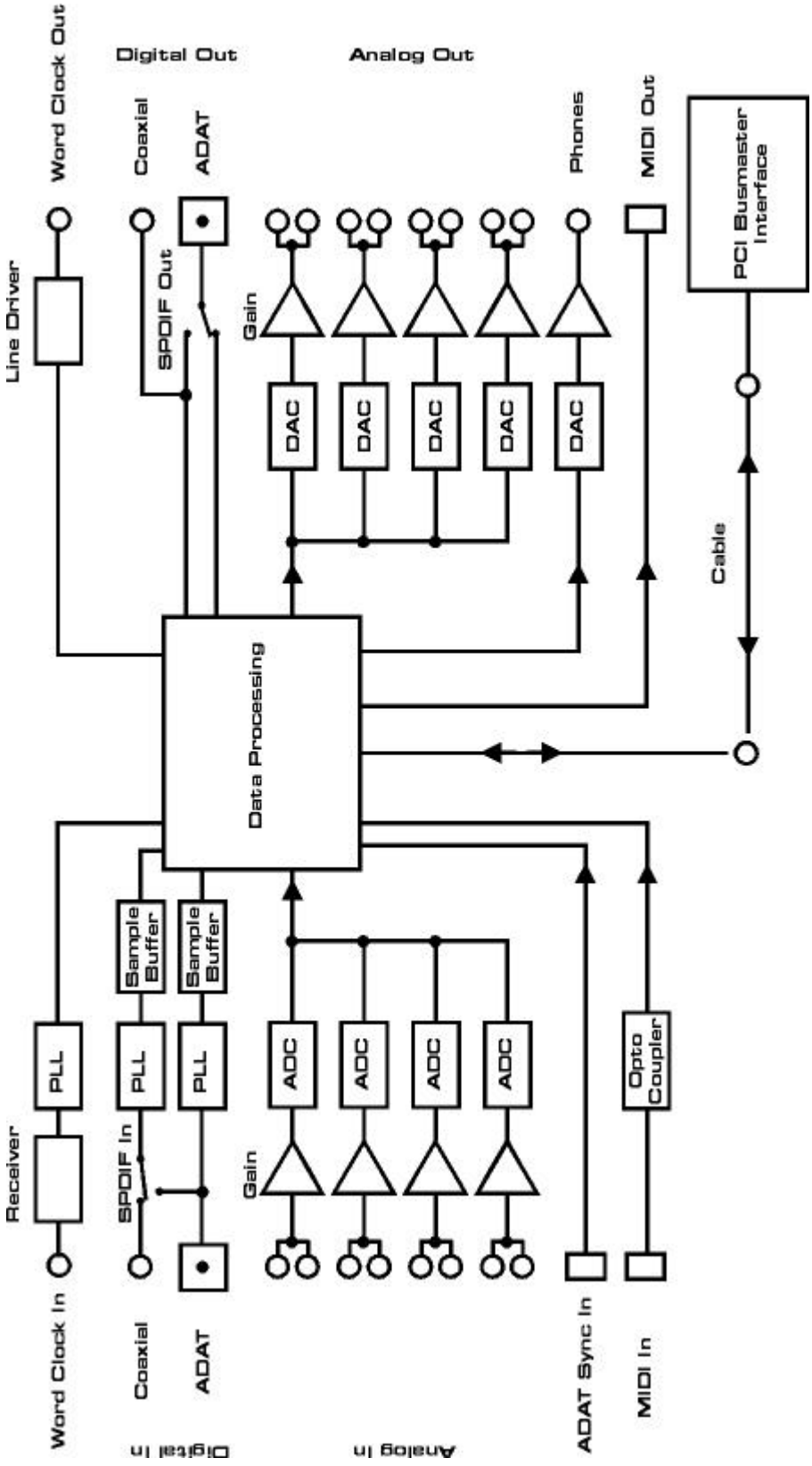
23. Diagrams

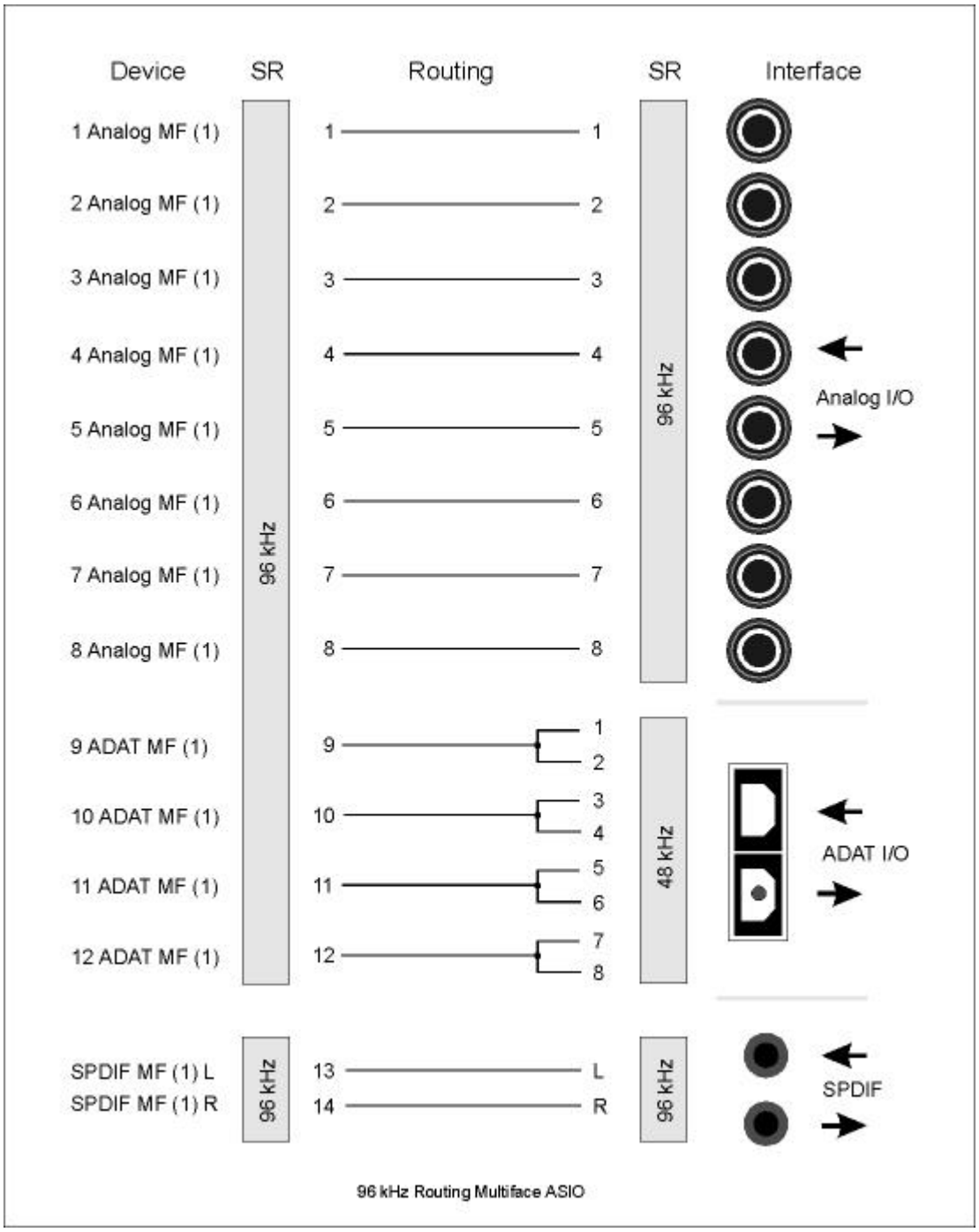
23.1 Block Diagram Multiface



23.2 ADAT Track Routing, ASIO at 96 kHz

This diagram shows the signal paths in ASIO double speed mode (88.2 / 96 kHz). The devices available under ASIO have been implemented according to the hardware. Signal routing is identical for record and playback.

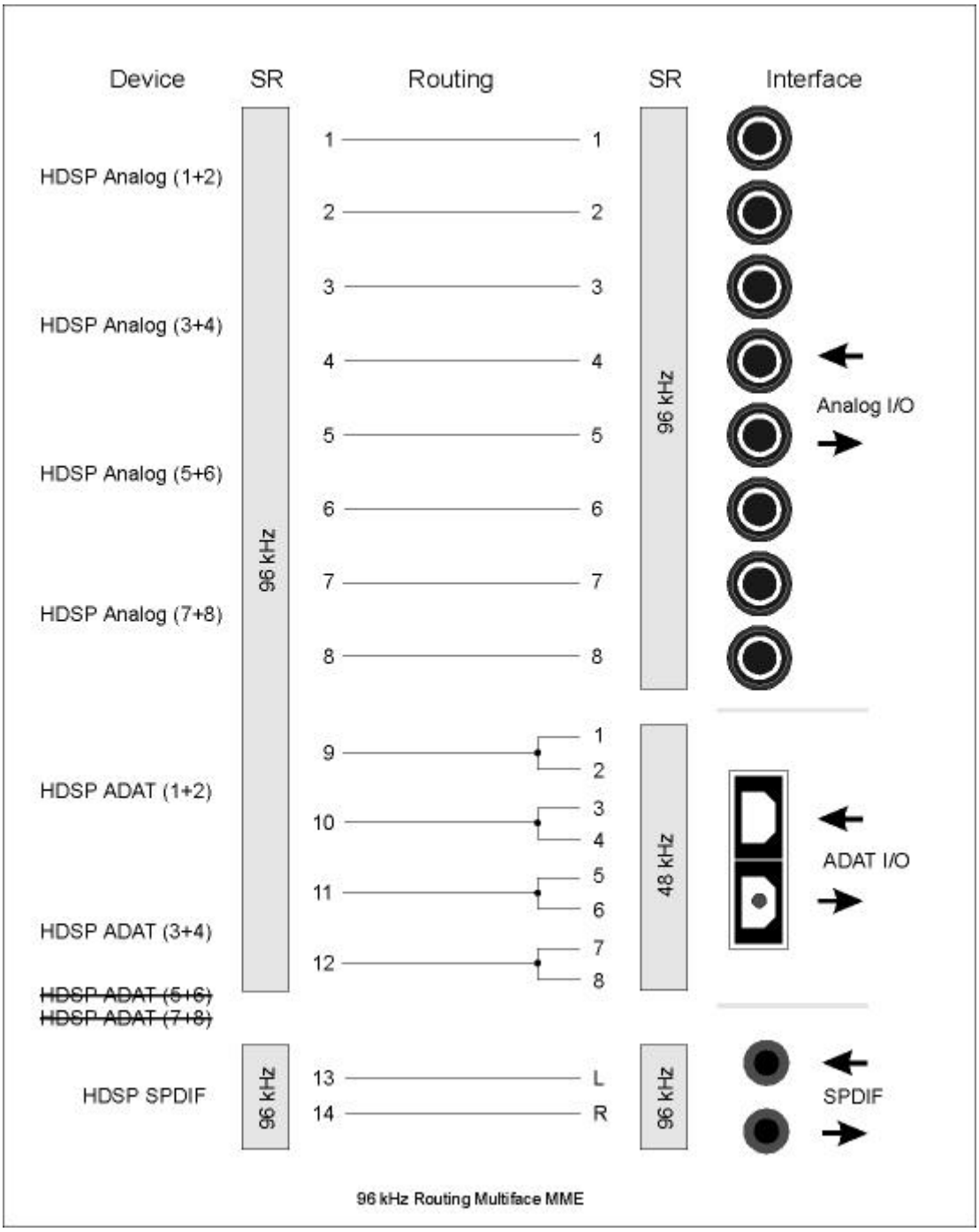
Device: The device name in the audio application SR: Sample Rate
Device name code: Channel in ASIO host, interface, Multiface, card number



23.3 ADAT Track Routing, MME at 96 kHz

This diagram shows the signal paths in MME double speed mode (88.2 / 96 kHz). The devices available via wave driver have been designed to avoid conflicts in normal operation, which is why channels 5, 6, 7 and 8 of the ADAT device have been omitted. Signal routing is identical for record and playback.

Device: The device name in the audio application SR: Sample Rate

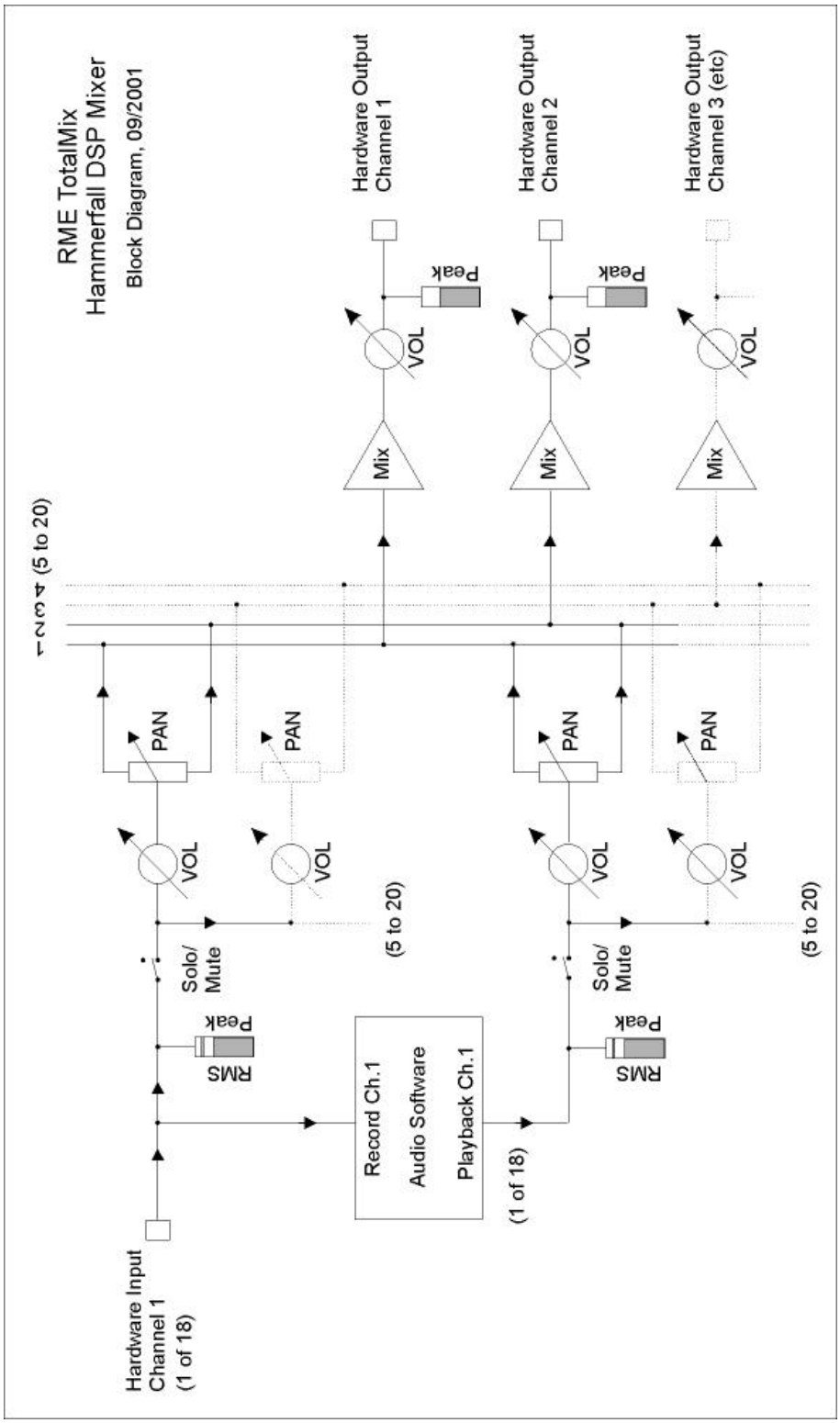


96 kHz Routing Multiface MME

23.4 Block diagram TotalMix

This diagram shows the signal flow inside the TotalMix mixer of the Multiface. It shall clarify the following function:

- The input signal of the hardware (ADAT/SPDIF In) is always directly fed through to the recording software. At the same time it can be routed to all 20 hardware outputs (Analog/ ADAT/ SPDIF/Monitor), even to all of them simultaneously.



24. CE and FCC Compliance Statements

CE

This device has been tested and found to comply with the EN55022 class B and EN50082-1 norms for digital devices, according to the European Council directive on counterpart laws in the member states relating to electromagnetic compatibility (EMVG).

FCC

This device has been tested and found to comply with the requirements listed in FCC Regulations, part 15 for Class 'B' digital devices. Compliance with these requirements provides a reasonable level of assurance that your use of this product in a residential environment will not result in harmful interference with other electronic devices.

This equipment generates radio frequencies and, if not installed and used according to the instructions in the User's Guide may cause interference harmful to the operation of other electronic devices.

Compliance with FCC regulations does not guarantee that interference will not occur in all installations. If this product is found to be the source of interference, which can be determined by turning the unit off and on again, please try to eliminate the problem by using one of the following measures:

- Relocate either this product or the device that is being affected by the interference
- Use power outlets on different branch circuits, or install AC line filters
- Contact your local retailer or any qualified radio and television engineer

When connecting external devices to this product, compliance to limits for a Class 'B' device requires the use of shielded cables.

FCC compliance statement: Tested to comply with FCC standards for home or office use.